DETAILED ACTION

Response to Arguments

In response to the applicant's argument as in the independent claim 1, wherein "applying the input signal and the filtered signal to a transformation circuit of the decorrelator and performing a matrixing operation on the input signal and the filtered signal to transform the input signal and the filtered signal into the first and second output signals, where the matrixing operation employs the correlation parameter and the level parameter" has been analyzed and is non-persuasive.

Since, Irwan et al. (US 2002/0118840 A1) specifically disclosed, of such feature wherein "applying the input signal and the filtered signal to a transformation circuit of the decorrelator and performing a matrixing operation on the input signal and the filtered signal to transform the input signal and the filtered signal into the first and second output signals, where the matrixing operation employs the correlation parameter and the level parameter (fig.2-3 (22,24-25); par [0024,027; 0037]/input signal and filtered input to a circuit as in (25) and matrix operation employ the correlation and strength/level parameter as being represented as beta and alpha, see fig.7, then output the audio channels as noted).

Similarly, as in claim 12; since, in regard to the applicant's argument of the examiner's conclusory statement by the examiner for the obviousness rejection has been further analyzed and is non-persuasive.

Since, having, the combined teaching of Irwan et al. and Ali as a whole, disclosed of wherein applying the input signal to the all- pass filter to generate the filtered signal comprises applying the input signal to the all- pass filter wherein the all-pass filter provides a frequency-dependent delay element ((fig.3; col.5 line 65-col.6 line 5) and further having the delay at a certain frequency is less than a delay at another frequency (fig.3, along graph as shown, the different frequencies as in (5K-8K Hz for example) having delay being less than other frequencies (0-4K Hz for example) as noted). Thus, it would have been obvious for one of the ordinary skills in the art to have tried in modifying the particular frequency is less than a delay at another frequency as disclosed with additionally implementing wherein the delay at a frequency Y is less than a delay at a frequency X, when Y > X with producing no unexpected result.

Similarly, a prior art has been provided in regard to the applicant's challenge in regard to claim 13 for the rejection of the official notice of the well known feature.

THIS ACTION IS MADE FINAL. Applicant is reminded of the extension of time policy as set forth in 37 CFR 1.136(a).

A shortened statutory period for reply to this final action is set to expire THREE MONTHS from the mailing date of this action. In the event a first reply is filed within TWO MONTHS of the mailing date of this final action and the advisory action is not mailed until after the end of the THREE-MONTH shortened statutory period, then the shortened statutory period will expire on the date the advisory action is mailed, and any extension fee pursuant to 37 CFR 1.136(a) will be calculated from the mailing date of the advisory action. In no event, however, will the statutory period for reply expire later than SIX MONTHS from the mailing date of this final action.

Claim Rejections - 35 USC § 101

2. 35 U.S.C. 101 reads as follows:

> Whoever invents or discovers any new and useful process, machine, manufacture, or composition of matter, or any new and useful improvement thereof, may obtain a patent therefor, subject to the conditions and requirements of this title.

Claims 20-22 are rejected under 35 U.S.C. 101 because as not falling within one of the four statutory categories of invention. The statutory "process" under 35 U.S.C. 101 must (1) be tied to another statutory category (such as a particular

apparatus), or (2) transform underlying subject matter (such as an article or material) to a different state or thing.

While the instant claim recites a series of steps or acts to be performed, such as "employing, and processing" with regard to claim 20, *the claim neither transforms underlying subject matter nor positively ties to another statutory category that accomplishes the claimed method steps, and may be interpreted as being done by the user without any use of a machine and therefore does not qualify as a statutory process.*

Allowable Subject Matter

a. Claims 14, 19, 22 are objected to as being dependent upon a rejected base claim, but would be allowable if rewritten in independent form including all of the limitations of the base claim and any intervening claims.

Claim Rejections - 35 USC § 102

3. The following is a quotation of the appropriate paragraphs of 35 U.S.C. 102 that form the basis for the rejections under this section made in this Office action:

> A person shall be entitled to a patent unless –
>
> ((a) the invention was known or used by others in this country, or patented or described in a printed publication in this or a foreign country, before the invention thereof by the applicant for a patent.

4. Claims 1-4, 7-8; 20 are rejected under 35 U.S.C. 102(a) as being anticipated by Irwan et al. (US 2002/0118840 A1).

RE claim 1, Irwan et al. disclose of a decorrelator, a method of synthesizing a first and a second output signal from an input signal, the method comprising: applying the input signal to a filter of the decorrelator to generate a filtered signal (fig.2 (24); fig.3 (24a, 24 b) ; par[0024]/ the input signal is filter and thereby produced a filtered input signal); and obtaining from an analysis circuit of the decorrelator, a correlation parameter indicative of a desired correlation between the first and second output signals (fig.3 (23); fig.5-7; par [0026, 0030-0035]/the transformation coefficient in which to map the output channels as desired again have the correlation parameter as in beta) and obtaining from the analysis circuit of the decorrelator a level parameter indicative of a desired level difference between the first and second output signals ((fig.3 (23); fig.5-7; par [0026, 0030-0035];and par [0044]/*strength of the signals* or level may also be determined and also see fig.7 the level difference of the output channel being represented by alpha) and applying the input signal and the filtered signal to a transformation circuit of the decorrelator and performing a matrixing operation on the input signal and the filtered signal

to transform the input signal and the filtered signal into the first and second output signals, where the matrixing operation employs the correlation parameter and the level parameter (fig.2-3 (22,24-25); par [0024,027; 0037]/input signal and filtered input to a circuit as in (25) and matrix operation employ the correlation and strength/level parameter as being represented as beta and alpha, see fig.7, then output the audio channels as noted).

Re claim 2, the method according to claim 1, wherein the matrixing operation comprises a common rotation by a predetermined angle of the first and second output signals in a space spanned by the input signal and the filtered input signal and where the predetermined angle depends on the level parameter (fig.6-7; par [0030-0031; 0035]/angle with correlation depend with strength level of alpha as noted).

Re claim 3, the method according to claim 2, but, wherein the predetermined angle is selected to inherently maximize a total contribution of the input signal to the first and second output signals (fig.4 (26); fig.7; par [0030-0031; 0044])/to determine the projection of the output channels and intensity as selected by the user as desired as in fig.7).

Re claim 4, the method according to claim 1, wherein further comprising scaling each of the first and second output signals to said desired level difference between the first and second output signals (fig.4 (26); par [0027, 0030-0031; 0044]/signal factor/strength for the channel is selectable by the user in determining the projection of the output channel output as in fig.7 based on coefficient characteristics).

Re claim 7, Irwan et al. disclose of a device for synthesizing a first and a second output signal from an input signal, the arrangement comprising: a filter for filtering the input signal to generate a filtered signal ((fig.2 (24); fig.3 (24a, 24 b) ; par[0024]/ the input signal is filter and thereby produced a filtered input signal); an analyzer for obtaining a correlation parameter indicative of a desired correlation between the first and second output signals and for obtaining a level parameter indicative of a desired level difference between the first and second output signals (fig.3 (23); fig.5-7; par [0026, 0030-0035]/the transformation coefficient in which to map the output channels as desired and having beta and alpha as the correlation and level parameters); a transformation circuit for transforming

the input signal and the filtered signal by a matrixing operation into the first and second output signals, where the matrixing operation depends on the correlation parameter and the level parameter (fig.2-3 (22,24-25); par [0024,027; 0037]/input signal and filtered input to a circuit as in (25) and matrix operation employ the correlation and strength/level parameter as being represented as beta and alpha, see fig.7, then output the audio channels as noted).

Re claim 8, the device of claim 7, further comprising an input unit for receiving an encoded audio signal (par [0004-0005]/system to be encoded for CD format) and a decoder for decoding the encoded audio signal to produce the input signal (fig.7, 8 (25); par [0004-0005]/system to be used on CD/to be decoded such encode signals so as to generate the audible surround sound signal to the listener).

Claim 20, Irwan et al. disclose of a data processing system, a method of synthesizing a first and a second output signal from an input signal, the method comprising: employing processing means of the data processing system to filter the input signal to generate a filtered signal (fig.2 (24); fig.3 (24a, 24 b) ;

par[0024]) and employing the processing means to obtain a correlation parameter indicative of a desired correlation between the first and second output signals, and to obtain a level parameter indicative of a desired level difference between the first and second output signals (fig.3 (23); fig.5-7; par [0026, 0030-0035]/the transformation coefficient in which to map the output channels as desired and having beta and alpha as the correlation and level parameters) and employing the processing means to perform a matrixing operation on the input signal and the filtered signal to transform the input signal and the filtered signal into the first and second output signals, where the matrixing operation employs the correlation parameter and the level parameter (fig.2-3 (22,24-25); par [0024,027; 0037]/input signal and filtered input to a circuit as in (25) and matrix operation employ the correlation and strength/level parameter as being represented as beta and alpha, see fig.7, then output the audio channels as noted).

Claim Rejections - 35 USC § 103

5. The following is a quotation of 35 U.S.C. 103(a) which forms the basis for all obviousness rejections set forth in this Office action:

> (a) A patent may not be obtained though the invention is not identically disclosed or described as set forth in section 102 of this title, if the differences between the subject matter sought to be patented and the prior art are such that the subject matter as a whole would have been obvious at the time the

> invention was made to a person having ordinary skill in the art to which said subject matter pertains. Patentability shall not be negatived by the manner in which the invention was made.

6. Claims 5-6; 11-12; 15-16; 21 are rejected under 35 U.S.C. 103(a) as being unpatentable over Irwan et al. (US 2002/0118840 A1) and Ali (US 6,895,093 B1).

Re claim 5, the method according to claim 1 with input signal being filtered, But, Irwan et al. fail to disclose of the filtering as being an all-pass filtering. But, Ali disclose of a system wherein similar concept of filtering with the filter being of the all-pass filtering (fig.3; col.5 line 65-col.6 line 5) for purpose of obtaining good stereophonic signal decorrelation without affecting stereo perception. Thus, it would have been obvious for one of the ordinary skill in the art to have modified the prior art with implementing the filtering of the input signal comprises all-pass filtering for purpose of obtaining good stereophonic signal decorrelation without affecting stereo perception.

Re claim 6, the method according to claim 5, wherein the all-pass filter comprises a frequency-dependant delay (col.5 line 65-col.6 line 5).

RE claim 11 has been analyzed and rejected with respect to claim 5.

Re claim 12, the method of claim 11, the combined teaching of Irwan et al. and Ali as a whole, disclosed of wherein applying the input signal to the all- pass filter to generate the filtered signal comprises applying the input signal to the all-pass filter wherein the all-pass filter provides a frequency-dependent delay element ((fig.3; col.5 line 65-col.6 line 5) and further having the delay at a certain frequency is less than a delay at another frequency (fig.3, along graph as shown, the different frequencies as in <u>(5K-8K Hz for example)</u> having delay being less than other frequencies <u>(0-4K Hz for example)</u> as noted).

Thus, it is noted it would have been obvious for one of the ordinary skills in the art to have tried in modifying the particular frequency is less than a delay at another frequency as disclosed with additionally implementing wherein the delay at a frequency Y is less than a delay at a frequency X, when Y > X with producing no unexpected result.

Re claims 15-16 have been analyzed and rejected with respect to claims 11-13 respectively.

Re claim 21 has been analyzed and rejected with respect to claim 11 respectively.

6. Claims 13; 17 are rejected under 35 U.S.C. 103(a) as being unpatentable over Irwan et al. (US 2002/0118840 A1) and Ali (US 6,895,093 B1) and Schroeder (IEEE, 1968).

Re claim 13, the method of claim 11 with the applying the input signal to the all- pass filter to generate the filtered signal comprises applying the input signal to the all- pass filter, but, the combined teaching of Irwan and Ali as a whole, fail to disclose of the specific wherein the all pass filter comprising one period of a Schroeder-phase complex. But, Schroeder disclose of such a concept wherein having such an all pass filter comprising one period of a Schroeder-phase complex (page 86-89; fig.1-4/filter with delay for a one T period signal) so as to minimize peak-to-peak amplitude and thereby yielding a low peak factor. Thus, it would have been obvious for one of the ordinary skill in the art to have modified the prior art with

implementing the all pass filter comprising one period of a Schroeder-phase complex so as to minimize peak-to-peak amplitude and thereby yielding a low peak factor.

Re claim 17 has been analyzed and rejected with respect to claim 13 respectively.

7. Claim 18 is rejected under 35 U.S.C. 103(a) as being unpatentable over Irwan et al. (US 2002/0118840 A1) and Shaffer et al. (US 6,973,184 B1).

Re claim 18, the device of claim 7, wherein the means for obtaining the correlation parameter and the means for obtaining the level parameter comprise an analysis circuit that receives a set of spatial parameters pertaining to the input signal including at least: an interaural phase difference (IPD) parameter and a value of a cross-correlation function parameter, and extracts from the set of spatial parameters the correlation parameter and the level parameter (par [0026; 0027; 00320035-0036], fig.6-7/phase parameter and correlation of the input signals as in y=y (k) –yb (k) and phase difference).

But, Irwan et al. fail to disclose of the parameters including: an interaural level difference (ILD) parameter; at least one of an interaural time difference (ITD) parameter.

But, Shaffer et al. disclose of a system wherein the parameters including: an interaural level difference (ILD) parameter; at least one of an interaural time difference (ITD) parameter (col.4 line 35-67; col.5 line 40-67). Thus, it would have been obvious for one of the ordinary skill in the art to have modified the combination with the parameters including: an interaural level difference (ILD) parameter; at least one of an interaural time difference (ITD) parameter for obtaining directional cues of the produced sound.

Similarly, it would have been obvious for one of the ordinary skills in the art to have tried in modifying the cross-correlation parameter with additionally implementing a maximum value of a cross-correlation function parameter with having no unexpected result produced so as to similarly obtain a directional cue of the produced sound as previously disclosed.

Conclusion

Any inquiry concerning this communication or earlier communications from the examiner should be directed to DISLER PAUL whose telephone number is (571)270-1187. The examiner can normally be reached on 7:30-5:00.

If attempts to reach the examiner by telephone are unsuccessful, the examiner's supervisor, Chin Vivian can be reached on 571-272-7848. The fax phone number for the organization where this application or proceeding is assigned is 571-273-8300.

Information regarding the status of an application may be obtained from the Patent Application Information Retrieval (PAIR) system. Status information for published applications may be obtained from either Private PAIR or Public PAIR. Status information for unpublished applications is available through Private PAIR only. For more information about the PAIR system, see http://pair-direct.uspto.gov. Should you have questions on access to the Private PAIR system, contact the Electronic Business Center (EBC) at 866-217-9197 (toll-free). If you would like assistance from a USPTO Customer Service Representative or access to the automated information system, call 800-786-9199 (IN USA OR CANADA) or 571-272-1000.

/D. P./
Examiner, Art Unit 2614

/Xu Mei/
Primary Examiner, Art Unit 2614